

User Manual

HandyTone-486

Analog Telephone Adaptor

For Firmware Version 1.0.6.2



Grandstream Networks, Inc.

www.grandstream.com

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1 Welcome

Congratulations on becoming an owner of HandyTone-486. You made an excellent choice and we hope you will enjoy all its capabilities.

Grandstream's award-winning HandyTone-486 is an all-in-one VoIP integrated access device that features superb audio quality, rich functionalities, high level of integration, compactness and ultra-affordability. The HandyTone-486 is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

Grandstream HandyTone-486 has been awarded the Best of Show product in 2004 Internet Telephony Conference and Expo.

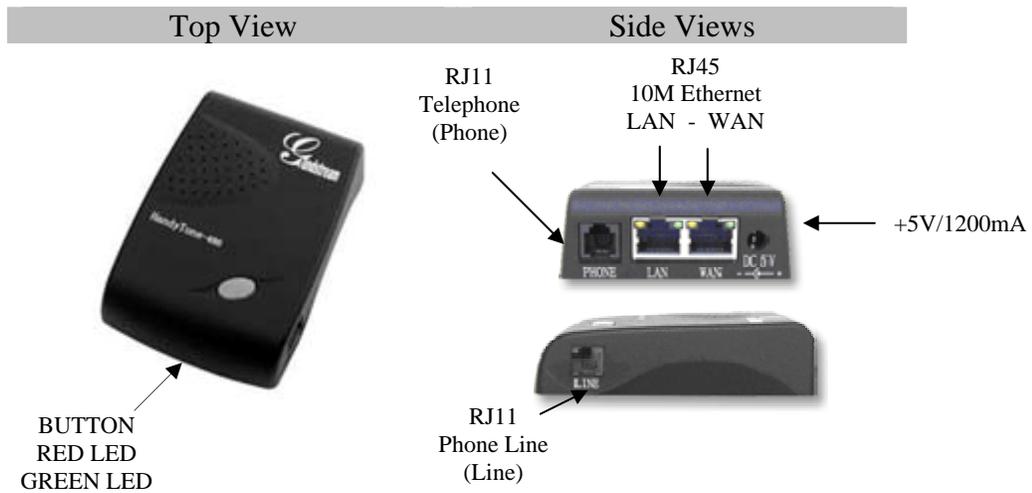


2 Installation

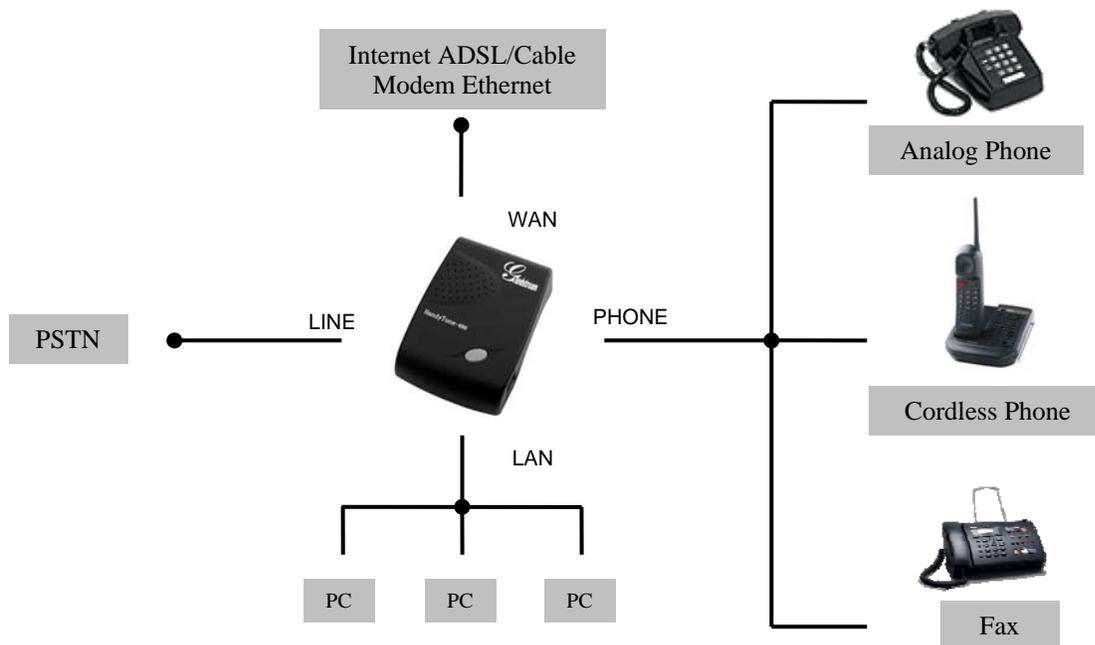
HandyTone-486 Analog Telephone Adaptor is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services.

The HandyTone-486 VoIP functionalities are available via a regular analog telephone.

The following photo illustrates the appearance of a HandyTone-486.



Interconnection Diagram of the HandyTone-486:



3 What is Included in the Package

The HandyTone-486 package contains:

- 1) One HandyTone-486
- 2) One universal power adaptor
- 3) One Ethernet cable

3.1 Safety Compliances

The HandyTone-486 is compliant with various safety standards including FCC/CE and C-tick. Its power adaptor is compliant with UL standard. The HandyTone-486 should only operate with the universal power adaptor provided in the package.

Warning: Please do not attempt to use a different power adaptor. Using other power adaptor may damage the HandyTone-486 and will void the manufacturer warranty.

Caution: Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

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4 Product Overview

4.1 Key Features

- Supports SIP 2.0(RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP (both client and server), NTP, PPPoE, STUN, TFTP, etc.
- Built-in router, NAT, Gateway and DMZ port forwarding. Can also be configured to function as a bridge (with disabled NAT and DHCP)
- Powerful digital signal processing (DSP) to ensure superb audio quality; advanced adaptive jitter control and packet loss concealment technology
- Support various codecs including G.711 (PCM a-law and u-law), G.723.1 (5.3K/6.3K), G.726 (40K/32K/24K/16K), as well as G.728, G.729 and iLBC.
- Support Caller ID/name display or block, Call waiting caller ID, Hold, Call Waiting/Flash, Call Transfer, Call Forward, in-band and out-of-band DTMF, Dial Plans, etc.
- Support fax pass through (for PCMU and PCMA) and T.38 FoIP (Fax over IP).
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Support standard encryption and authentication (DIGEST using MD5 and MD5-sess)
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support automated NAT traversal without manual manipulation of firewall/NAT
- Support device configuration via built-in IVR, Web browser or Central configuration files through TFTP or HTTP server
- Support firmware upgrade via TFTP or HTTP with encrypted configuration files.
- Support PSTN pass through, be able to make and receive VoIP or PSTN calls using same connected analogue phone.
- Ultra compact (wallet size) and lightweight design, great companion for travelers.
- Compact, lightweight Universal Power adapter.

4.2 Hardware Specification

The table below describes the features of the model.

<u>Model</u>	<u>HandyTone-486</u>
LAN interface	1xRJ45 10Base-T
WAN interface	1xRJ45 10Base-T
FXS telephone port	1xFXS
PSTN port	1x PSTN pass through or life line port
Button	1
LED	Green and red color
Universal Switching Power Adaptor	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA, UL certified
Dimension	70mm (W) 130mm (D) 27mm (H)
Weight	0.6lbs (0.3kg)
Temperature	40 - 130°F 5 - 45°C
Humidity	10% - 90% (non-condensing)
Compliance	

NOTES:

- *On the bottom of 486, there is a sticker. If there is Rev:2.0 on the right of tick sign, it means the HT486 is Rev:2.0 (or HT487 in nickname). The difference between HT486 (Rev:1.0) and HT486 Rev:2.0 is that HT486 Rev:2.0 line port is PSTN pass through while HT486 (Rev:1.0, old model, no longer shipped) line port is just life line port and will bridge to PSTN only when there is no power.*
- *The HT486 Rev:2.0 and HT486 (Rev:1.0) are hardware difference and can not change via firmware upgrade.*

5 Basic Operations

5.1 Get Familiar with Key Pad and Voice Prompt

HandyTone-486 has stored a voice prompt menu (Interactive Voice Response or IVR) for quick browsing and simple configuration.

To enter this voice prompt menu, simply **press the button or “****” from the analog phone.**

Menu	Voice Prompt	User's Options
Main Menu	“Enter a Main Option”	Enter ‘*’ to next option and ‘#’ back to main menu. Enter 01-06, 99 menu option
01	“DHCP Mode”, “Static IP Mode”	Enter ‘9’ to toggle the selection
02	“IP Address “ + IP address	It will prompt you with the current WAN IP address. Enter 12 digit new IP address if in Static IP Mode
03	“Subnet “ + IP address	Same as menu 02
04	“Gateway “ + IP address	Same as menu 02
05	“DNS Server “ + IP address	Same as menu 02
06	“TFTP Server “ + IP address	Same as menu 02
47	“Direct IP Calling”	When entered, you will be prompted a dial tone, then enter 12 digit IP address This menu can be also entered by pressing the button again
86	“Voice Messages Pending” “No Voice Messages”	Enter 9 to dial pre-configured phone number to retrieve VM
99	“RESET”	Enter ‘9’ to confirm the RESET Enter MAC address to restore factory default setting
	“Invalid Entry”	Automatically return to Main Menu

NOTES:

- Once the button is pressed, it enters the IVR main menu. If the button is pressed again, while it is already in the voice prompt menu state, it jumps to “Direct IP Calling” option and dial tone plays in this state to allow direct IP to IP calling
- ‘*’ functions similar to ‘↓’ key of BT-100 phone to select the next menu option
- ‘#’ returns back to main MENU
- ‘9’ is similar to ENTER key in many cases to confirm an option
- All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. Once all digits are accumulated, it will automatically process them
- For IP address input, ignore the dot and key in the digit directly, add 0 before octet with less than three digits. e.g.: IP: 192.168.1.10, key in: 192 168 001 010
- Key entry cannot be deleted but the phone may prompt error once it is detected

5.2 Make Phone Calls

5.2.1 Calling phone or extension numbers

There are currently two methods to make an extension number call:

- a) Dial the numbers directly and wait for 4 (default) seconds.
(this default parameter can be adjusted in the web configuration page)
- b) Dial the numbers directly, and press # (assuming that “use #” as dial key is selected in web configuration).

Other functions available during the call are call-waiting/flash, call transfer (both blind and consultant) and call forwarding, assume that an ITSP is used.

5.2.2 Direct IP calls

Direct IP calling allows two parties, that is, a HandyTone with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy. This kind of VoIP calls can be made between two parties if:

- both HandyTone ATA and other VoIP Device(i.e., another HandyTone ATA or Budgetone SIP phone or other VoIP unit) have public IP addresses, or
- both HandyTone ATA and other VoIP Device are on the same LAN using private or public IP addresses, or
- Both HandyTone ATA and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

To make a direct IP to IP call, first pick up the analog phone or turn on the speakerphone on the analog phone, then follow Section 4.1 with voice prompt 47, followed by the 12-digit target IP address. Destination ports can also be specified using *4 (encoding for “:”) followed by the encoded port number. Following is a table of the encoding scheme for the most commonly used characters:

Input	Encoding
00	0
01	1
02	2
03	3
04	4
05	5
06	6
07	7
08	8
09	9
*0	. (dot character)
*4	: (column character)

Examples:

If the target IP address is 192.168.0.160, the dialing convention is

Voice Prompt with option 47, then 192168000160

followed by pressing the “#” key if it is configured as a send key or wait for more than 5 seconds. In this case, the default destination port 5060 is used if no port is specified.

If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:

Voice Prompt with option 47, then 192168001020*45062 followed by pressing the “#” key if it is configured as a send key or wait for 5 seconds.

Currently, we don't support inter-domain IP to IP call like: john123@othervoip.com

5.2.3 Blind Transfer

Assume that call party A and B are in conversation. A wants to *Blind Transfer* B to C:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. Then A dials *87 then dials C's number, and then #(or wait for 4 seconds)
3. A can hang up.

NOTES:

- *Call Feature has to be set to YES in web configuration page.*

A can hold on to the phone and await one of the three following behaviors:

- A quick confirmation tone (temporarily using the call waiting indication tone) follows by a dial tone. This indicates the transfer has been successful (transferee has received a 200 OK from transfer target). At this point, you can either hang up or make another call.
- A quick busy tone followed by a restored call (On supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
- Busy tone keeps playing. This means we have failed to receive the second NOTIFY from the transferee and decide to time out. Note: this does not indicate the transfer has been successful, nor does it indicate the transfer has failed. When transferee is a client that does not support the second NOTIFY, this will be the case. In bad network scenarios, this could also happen, although the transfer may have been completed successfully.

5.2.4 Attended Transfer

Assume that call party A and B are in conversation. A wants to *Attend Transfer* B to C:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone
2. A then dial C's number then # (or wait for 4 seconds), A and C now in conversation.
3. Then A can hang up.

NOTES:

- *When intended Transfer failed, if A hangs up, the HandyTone 486 will ring user A back again to remind A that B is still on the call, by pressing FLASH or Hook again will restore conversation between A and B.*

5.2.5 Send and Receive PSTN Calls Through PSTN Line Port

Starting with HandyTone 486 Rev:2.0, user will be able to send and receive calls from PSTN. To receive PSTN calls, simply make phone off hook when the analog phone rings. To make a PSTN call, simply press *00 (default, or any number configured in web configuration page) to switch to the PSTN line and get dial tone and call out using the PSTN line.

5.3 Call Features

5.3.1 Call Features Table (star code)

Following table shows the call features (* code) of HandyTone-486.

Key	Call Features
*30	Block CallerID (for all-config change)
*31	Send CallerID (for all-config change)
*67	Block CallerID (per call)
*82	Send CallerID (per call)
*50	Disable Call Waiting (for all-config change)
*51	Enable Call Waiting (for all-config change)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward. To use this feature, dial “*72”, wait for the dial tone. Then dial the forward number, wait for dial tone, hang up.
*73	Cancel Unconditional Call Forward To cancel “Unconditional Call Forward”, dial “*73” and get the dial tone, then hang up.
*90	Busy Call Forward To use this feature, dial “*90”, wait for the dial tone. Then dial

	the forward number, wait for dial tone and hang up.
*91	Cancel Busy Call Forward To cancel “Busy Call Forward”, dial “*91” and get the dial tone, then hang up
*92	Delayed Call Forward To use this feature, dial “*92”, wait for the dial tone. Then dial the forward number, wait for dial tone, hang up.
*93	Cancel Delayed Call Forward To cancel this Forward, dial “*93” and get the dial tone, then hang up
Flash/Hook	When in conversation, this action will switch to the new incoming call if user heard the call waiting sound. When in conversation and no incoming call heard, this action will switch to a new channel for a new call.

5.3.2 PSTN Pass Through / Life Line

When HandyTone-486 is out of power, the RJ 11 line jack on the HandyTone-486 side will function as a pass through jack. The user will be able to use the same analog phone for PSTN calls.

5.4 LED Light Pattern Indication

Following are the LED light pattern indication.

RED LED always indicates abnormal status	
DHCP Failed or WAN No Cable	flash every 2 seconds (if DHCP is configured)
HandyTone-486 fails to register	flash every 2 seconds (if SIP server is configured)

GREEN LED always for normal working status	
Message Waiting Indication	Button flashes every 2 seconds
RINGING	Button flashes at 1/10 second
RINGING INTERVAL	Button flashes every second

6 Configuration Guide

6.1 Configuring HandyTone-486 WAN IP through Voice Prompt

6.1.1 DHCP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-486 to use DHCP.

6.1.2 STATIC IP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-486 to use STATIC IP mode, then use option 02, 03, 04 to set up HandyTone's IP, Subnet Mask, Gateway respectively.

6.2 Configuring HandyTone-486 with Web Browser

HandyTone-486 series ATA has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the HandyTone through a Web browser such as Microsoft's IE and AOL's Netscape.

6.2.1 Access the Web Configuration Menu

The HandyTone-486 HTML configuration menu can be accessed via LAN or WAN port:

- From the LAN port use the default LAN gateway IP address (this can be changed to any private LAN segment by user, just make sure WAN and LAN are using different LAN segment if HT486 WAN side is connected behind a SOHO router in SOHO router's LAN port):

http://192.168.2.1

- The WAN port HTML configuration menu option is **disabled** from factory. To access the HTML configuration menu from the WAN port, first enable the "WAN side HTTP access" option by accessing the configuration via HT486's LAN port. With the WAN side HTTP access enable, then get the WAN IP address of the HandyTone through IVR (section 5.1 with menu option 02). Then the HandyTone's Web Configuration Menu can be accessed by the following URI via WAN port:

http://HandyTone-IP-Address,

where the ***HandyTone-IP-Address*** is the WAN IP address of the HandyTone-486.

NOTE:

- *If typing IP address into address of browser to get into the configuration page, please strip out the “0” announced in IVR or displayed in the phone if using BT or GXP phones as the browser will parse in octet. e.g.: If IVR announce the IP address: 192.168.001.014, please just key in IP address into the address field by: 192.168.1.14. Otherwise if you key in 192.168.001.014 into browser you are actually try to talk to: 192.168.1.12 since the browser parse it in octet.*

6.2.2 End User Configuration

Once this request is entered and sent from a Web browser, the HandyTone-486 will respond with the following login screen:

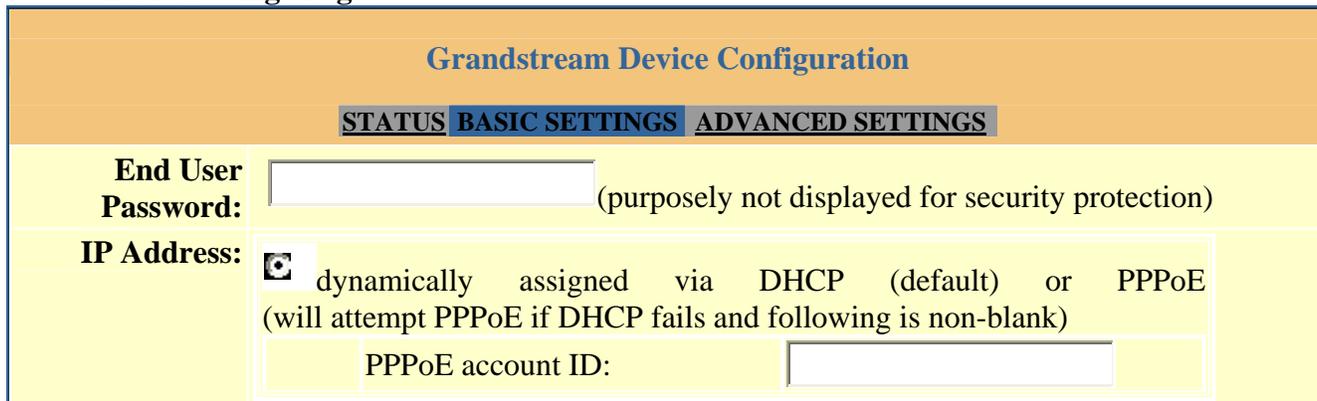


The password is case sensitive with maximum length of 25 characters. The factory default password for End User is “123” or just blank; for Administrator is “admin”. Only administrator can get access to “ADVANCED SETTING” configuration page.

After the correct password is entered in the login screen, the embedded Web server inside the HandyTone-486 will respond with the Configuration Menu page screen which is explained in details below.

Following are configuration pages End User will see once log in correctly:

- **Basic Settings Page:**



PPPoE password: <input type="text"/>	
Preferred DNS server: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>	
<input type="checkbox"/> statically configured as:	
IP Address:	<input type="text" value="192"/> <input type="text" value="168"/> <input type="text" value="1"/> <input type="text" value="10"/>
Subnet Mask:	<input type="text" value="255"/> <input type="text" value="255"/> <input type="text" value="255"/> <input type="text" value="0"/>
Default Router:	<input type="text" value="192"/> <input type="text" value="168"/> <input type="text" value="1"/> <input type="text" value="254"/>
DNS Server 1:	<input type="text" value="192"/> <input type="text" value="168"/> <input type="text" value="1"/> <input type="text" value="254"/>
DNS Server 2:	<input type="text" value="151"/> <input type="text" value="203"/> <input type="text" value="0"/> <input type="text" value="85"/>
Time Zone:	current setting is " GMT-5:00 (US Eastern Time, New York)" <input type="button" value="v"/>
Daylight Savings Time:	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if set to Yes, display time will be 1 hour ahead of normal time)
NAT/Bridge/DHCP Server Information & Configuration:	
<i>Device Mode:</i>	<input checked="" type="checkbox"/> NAT Router <input type="checkbox"/> Bridge
<i>Cloned WAN MAC Addr:</i>	<input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> (in hex format)
<i>LAN Subnet Mask:</i>	<input type="text" value="255.255.255.0"/> (default is 255.255.255.0)
<i>LAN DHCP Base IP:</i>	<input type="text" value="192.168.2.1"/> (base IP for the LAN port, default is 192.168.2.1)
<i>DHCP IP Lease Time:</i>	<input type="text" value="120"/> (in units of hours, default is 120 hours or 5 days)
<i>DMZ IP:</i>	<input type="text"/>
<i>Port Forwarding:</i>	WAN port <input type="text" value="0"/> LAN IP <input type="text"/> LAN port <input type="text" value="0"/> Protocol <input type="text" value="' TCP & UDP'"/>
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	WAN port <input type="text" value="0"/> LAN IP <input type="text"/> LAN port <input type="text" value="0"/> Protocol <input type="text" value="' TCP & UDP'"/>
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>	

- **Status Page:**

Grandstream Device Configuration	
STATUS BASIC SETTINGS ADVANCED SETTINGS	
MAC Address:	00.0B.82.01.FC.52
WAN IP Address:	192.168.1.10
Product Model:	HT487
Software Version:	Program-- 1.0.6.2 Bootloader-- 1.0.1.0 HTML-- 1.0.0.48 VOC-- 1.0.0.9
System Up Time:	7 day(s) 2 hour(s) 40 minute(s)
Registered:	Yes
PPPoE Link Up:	disabled
NAT:	detected NAT type is open Internet
NAT Mapped IP:	0.0.0.0
NAT Mapped Port:	0
Total Inbound Calls:	0
Total Outbound Calls:	0
Total Missed Calls:	0
Total Call Time (in minutes):	0
Total SIP Message Sent:	30801
Total SIP Message Received:	31039
Total RTP Packet Sent:	0
Total RTP Packet Received:	0
Total RTP Packet Loss:	0
All Rights Reserved Grandstream Networks, Inc. 2005	

Here are explanations of the above two pages:

MAC Address	The device ID, in HEX format. This is very important ID for ISP troubleshooting.
WAN IP Address	This field shows WAN port IP address.
Product Model	This field contains the product model info, like HT487 is HT486 Rev:2.0
Software Version	<p>Program: This is the main software release. <i>This number is always used for firmware upgrade.</i> Here the box showed: 1.0.6.0</p> <p>Bootloader: This is normally not changed.</p> <p>HTML: This is the user interface, normally not changed.</p> <p>VOC: This is the codec program, normally not changed.</p>

<i>System Uptime</i>	This shows system up time since last reboot.
<i>Registered</i>	This shows whether the unit is registered to service provider's server.
<i>PPPoE Link Up</i>	This shows whether the PPPoE is up if connected to DSL modem
<i>NAT</i>	This shows what kind NAT the HandyTone is connected to via its WAN port. It is based on STUN protocol.
<i>NAT Mapped IP</i>	WAN side public IP if connected to LAN of a SOHO router.
<i>Other Statistical Status of ATA</i>	Self explainable, see the page displayed.
BASIC SETTINGS:	
<i>End User Password</i>	This contains the password to access the Web Configuration Menu. This field is case sensitive with max. 25 characters
<i>IP Address</i>	<p>There are 2 modes where the HandyTone ATA can operate:</p> <ul style="list-style-type: none"> - If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The HandyTone ATA will acquire its IP address from the first DHCP server it discovers from the LAN it is connected. <p>To use PPPoE feature set the PPPoE account settings here if connect the box to DSL modem. The HandyTone will attempt to establish a PPPoE session if any of the PPPoE fields is set. In this mode, the WAN side web access is disabled and TFTP upgrade for firmware is not feasible and HTTP upgrade is only available solution.</p> <ul style="list-style-type: none"> - If Static IP mode is selected, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured. These fields are reset to zero by default.
<i>Time Zone</i>	This parameter controls how the displayed date/time will be adjusted according to the specified time zone.
<i>Daylight Savings Time</i>	This parameter controls whether the displayed time will be daylight savings time or not. If set to Yes, then the displayed time will be 1 hour ahead of normal time.
<i>Device Mode</i>	This parameter controls whether the device is working in NAT router mode or Bridge mode. Need save the setting and reboot the device before the setting start to work

Cloned WAN MAC Address:	Allow the user to set a specific MAC address. Set in Hex format
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP:	Base IP for the LAN port, which function as a Gateway for the subnet. Default value is 192.168.2.1
DHCP IP Lease Time:	Value is set in units of hours. Default value is 120hr (5 Days.) The time IP address are assigned to the LAN clients
DMZ IP:	Forward all WAN IP traffic to a specific IP address if no matching port is used by HandyTone-486 itself or in the defined port forwarding.
Port Forwarding:	Allow the user to forward a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port.

6.2.3 Advanced User Configuration

To login to the Advanced User Configuration page, follow the instruction in section 6.2.1 to get to the following login page. The password is case sensitive and the factory default password for Advanced User is 'admin'.

Advanced User configuration page includes not only the end user configuration, but also some advanced configuration such as SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configurations. Following is the snap shot of the configuration page.

Grandstream Device Configuration	
STATUS BASIC SETTINGS ADVANCED SETTINGS	
Admin Password:	<input type="text"/> (purposely not displayed for security protection)
SIP Server:	<input type="text" value="my.voipprovider.com"/> (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	<input type="text" value="my.voipprovider.com:port"/> (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	<input type="text" value="1234567"/> (the user part of an SIP address)
Authenticate ID:	<input type="text" value="1234567"/> (can be identical to or different from SIP User ID)
Authenticate Password:	<input type="text"/> (purposely not displayed for security protection)
Name:	<input type="text" value="Your Name Here"/> (optional, e.g., John Doe)
Advanced Options:	
<i>Preferred Vocoder:</i> (in listed order)	choice 1: <input pcmu"="" type="text" value="current setting is "/> choice 2: <input pcma"="" type="text" value="current setting is "/> choice 3: <input g723"="" type="text" value="current setting is "/> choice 4: <input g729"="" type="text" value="current setting is "/> choice 5: <input g726-32"="" type="text" value="current setting is "/> choice 6: <input g728"="" type="text" value="current setting is "/> choice 7: <input ilbc"="" type="text" value="current setting is "/>
G723 rate:	<input checked="" type="checkbox"/> 6.3kbps encoding rate <input type="checkbox"/> 5.3kbps encoding rate
iLBC frame size:	<input checked="" type="checkbox"/> 20ms <input type="checkbox"/> 30ms
iLBC payload type:	<input type="text" value="97"/> (between 96 and 127, default is 99)
Silence Suppression:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
Voice Frames per TX:	<input type="text" value="2"/> (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Fax Mode:	<input checked="" type="checkbox"/> T.38 (Auto Detect) <input type="checkbox"/> Pass-Through
Layer 3 QoS:	<input type="text" value="48"/> (Diff-Serv or Precedence value)
Layer 2 QoS:	802.1Q/VLAN Tag <input type="text" value="0"/> 802.1p priority value <input type="text" value="0"/> (0-7)
Use DNS SRV:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
User ID is phone number:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes

<i>SIP Registration:</i>	<input checked="" type="checkbox"/> Yes <input type="checkbox"/> No
<i>Unregister On Reboot:</i>	<input checked="" type="checkbox"/> Yes <input type="checkbox"/> No
<i>Register Expiration:</i>	<input type="text" value="60"/> (in minutes, default 1 hour, max 45 days)
<i>Early Dial:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (use "Yes" only if proxy supports 484 response)
<i>Dial Plan Prefix:</i>	<input type="text"/> (this prefix string is added to each dialed number)
<i>No Key Entry Timeout:</i>	<input type="text" value="4"/> (in seconds, default is 4 seconds)
<i>Use # as Dial Key:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
<i>local SIP port:</i>	<input type="text" value="5060"/> (default 5060)
<i>local RTP port:</i>	<input type="text" value="5004"/> (1024-65535, default 5004)
<i>Use random port:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes
<i>NAT Traversal:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes, STUN server is: <input type="text"/> (URI or IP:port)
<i>keep-alive interval:</i>	<input type="text" value="20"/> (in seconds, default 20 seconds)
<i>Use NAT IP</i>	<input type="text"/> (if specified, this IP address is used in SIP/SDP message)
<i>Proxy-Require:</i>	<input type="text"/> (if specified, the content will appear in Proxy-Require header)
<i>Firmware Upgrade:</i>	<input checked="" type="checkbox"/> Via TFTP Server <input type="text" value="168"/> <input type="text" value="75"/> <input type="text" value="215"/> <input type="text" value="188"/> <input checked="" type="checkbox"/> Via HTTP Server <input type="text" value="fm.grandstream.com/gs"/> Automatic HTTP Upgrade: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes, check for upgrade every <input type="text" value="7"/> days (default 7 days)
<i>SUBSCRIBE for MWI:</i>	<input checked="" type="checkbox"/> No, do not send SUBSCRIBE for Message Waiting Indication <input type="checkbox"/> Yes, send periodical SUBSCRIBE for Message Waiting Indication
<i>Offhook Auto-Dial:</i>	<input type="text"/> (User ID/extension to dial automatically when offhook)
<i>Enable Call Features:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
<i>Disable Call-Waiting:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>Send DTMF:</i>	<input checked="" type="checkbox"/> in-audio <input type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO
<i>DTMF Payload Type:</i>	<input type="text" value="101"/>
<i>Send Flash Event:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (Flash will be sent as a DTMF event if set to Yes)
<i>FXS Impedance:</i>	current setting is " 600 Ohm (North America)" <input type="text"/>

<i>Caller ID Scheme:</i>	current setting is " Bellcore" 
<i>Onhook Voltage:</i>	current setting is " 36V" 
<i>Polarity Reversal:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (reverse polarity upon call establishment and termination)
<i>NTP Server:</i>	time.nist.gov (URI or IP address)
<i>Send Anonymous:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (caller ID will be blocked if set to Yes)
<i>Lock keypad update:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (configuration update via keypad is disabled if set to Yes)
<i>Special Feature:</i>	current setting is "Standard" 
<i>WAN Side HTTP Access:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (WAN side access to http server will be rejected if set to No)
<i>PSTN Access Code:</i>	<input type="text"/> (key pattern to use the PSTN line, default is "*00")
<i>Syslog Server:</i>	<input type="text"/>
<i>Syslog Level:</i>	current setting is "NONE" 
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>	
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<i>Admin Password</i>	Administrator password. Only administrator can configure the “Advanced Settings” page. Password field is purposely blanked for security reason after clicking update and saved. The maximum password length is 25 characters.
<i>SIP Server</i>	IP address or Domain name provided by VoIP service provider
<i>Outbound Proxy</i>	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by ATA for firewall or NAT penetration in different network environment. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy will provide solution for it.
<i>SIP User ID</i>	User account information, provided by VoIP service provider (ITSP), usually has the form of digit similar to phone number or actually a phone number
<i>Authenticate ID</i>	ID used for authentication, usually same as SIP user ID, but could be different and decided by ITSP.
<i>Authentication Password</i>	Account information, password for ATA to register to (SIP) servers of ITSP.
<i>Name</i>	Field for user to put in additional information to be displayed when calling, like caller ID.
<i>Preferred Vocoder</i>	The HandyTone ATA supports up to 7 different codec types including G.711 A/U law , G.723.1, G.726, G.728, G.729A/B, iLBC. Depending on the product model, some of these codecs may not be provided in standard release. A user can configure codecs in a preference list that will be included with the same preference order in SDP message. The first codecs in this list can be entered by choosing the appropriate option in “Choice 1”. Similarly, the last codecs in this list can be entered by choosing the appropriate option in “Choice 6”.

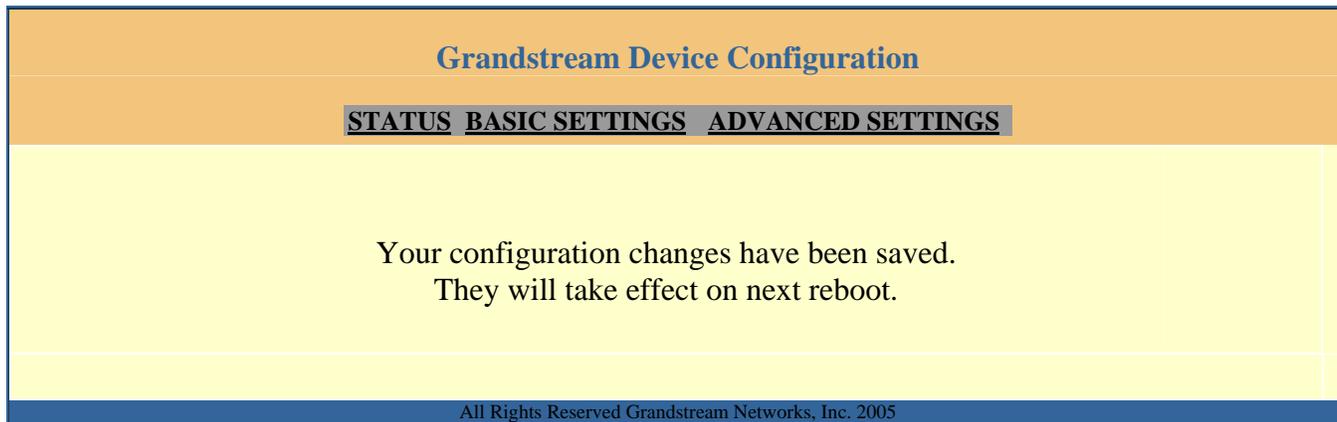
G723 Rate:	This defines the encoding rate for G723 codec. By default, 6.3kbps rate is chosen.
iLBC frame size:	This sets the iLBC packet frame size in 20ms or 30ms. Default is 20ms. For Asterisk PBX, 30ms might need to be set.
iLBC payload type:	This defines payload time for iLBC. Default value is 98. The valid range is between 96 and 127. For Asterisk PBX, 97 might need to be set.
Silence Suppression	This controls the silence suppression/VAD feature of G723 and G729. If set to “Yes”, when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to “No”, this feature is disabled.
Voice Frames per TX	<p>This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first codec in the above codec Preference List or the actual used payload type negotiated between the 2 conversation parties at run time.</p> <p>e.g., if the first codec is configured as G723 and the “Voice Frames per TX” is set to be 2, then the “ptime” value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first codec chosen is G729 or G711 or G726, then the “ptime” value in the SDP message of an INVITE request will be 20ms.</p> <p>If the configured voice frames per TX exceeds the maximum allowed value, the HandyTone ATA will use and save the maximum allowed value for the corresponding first codec choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively. Please be careful when message those parameters.</p>
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48.
Layer 2 QoS	This contains the value used for layer 2 VLAN tag. Default setting is blank.
Use DNS SRV:	Default is No. If set to Yes the client will use DNS SRV to lookup for the server
User ID is Phone Number	If the HandyTone ATA has an assigned PSTN telephone number, then this field will be set to “Yes”. Otherwise, set it to “No”. If “Yes” is set, a “user=phone” parameter will be attached to the “From” header in SIP request
SIP Registration	This parameter controls whether the HandyTone ATA needs to send REGISTER messages to the proxy server. The default setting is “Yes”.
Unregister on Reboot	Default is No. If set to yes, the SIP user will be unregistered on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the HandyTone ATA will refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Early Dial	Default is No. Use only if proxy supports 484 response
Dial Plan Prefix	Sets the prefix added to each dialed number
No Key Entry timeout	Default is 4 seconds. User can short or extend that depends on digits dialed

Use # as Send Key	This parameter allows the user to configure the “#” key to be used as the “Send”(or “Dial”) key. Once set to “Yes”, pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this key is essentially equivalent to the “(Re)Dial” key. If set to “No”, this # key will then be included as part of the dial string to be sent out.
Local SIP port	This parameter defines the local SIP port the HandyTone ATA will listen and transmit. The default value is 5060.
Local RTP port	This parameter defines the local RTP-RTCP port pair the HandyTone ATA will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port_value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value is 5004.
Use Random Port	This parameter, when set to Yes, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple HandyTone ATAs are behind the same NAT.
NAT Traversal	This parameter defines whether the HandyTone ATA NAT traversal mechanism will be activated or not. If activated (by choosing “Yes”) and a STUN server is also specified, then the HandyTone ATA will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the HandyTone ATA will attempt to detect if and what type of firewall/NAT it is behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the HandyTone ATA will attempt to use its mapped public IP address and port in all the SIP and SDP messages it sends out. If this field is set to “Yes” with no specified STUN server, then the HandyTone ATA will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the “hole” on the NAT open.
Keep-alive interval	The interval of sending blank UDP packet to keep NAT “pin hole” open, default is 20
Use NAT IP:	NAT IP address used in SIP/SDP message. Default is blank.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Firmware Upgrade: Via TFTP Server	This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, the HandyTone ATA will attempt to retrieve new configuration file or new code image from the specified TFTP server at boot time. It will make up to 3 attempts before timeout and then it will start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image will be verified and then saved into the Flash memory. <i>Be careful when the TFTP is in progress (the LED will flash in red fast). Do NOT interrupt the process (especially the power supply) otherwise will cause the device dead. Be patient as in some network this process will take more than 15 minutes.</i>
Via HTTP Upgrade Server	The URL for the web server used for firmware upgrade and configuration via HTTP. For example, http://provisioning.mycompany.com:6688/Grandstream/1.0.6.0 Here “:6688” is the specific TCP port that the HTTP server is listening to, it can be omitted if using default port 80. <i>Note: Auto Upgrade has to be set to Yes to enable HTTP upgrade. TFTP and HTTP options are mutual exclusive.</i>
Automatic Upgrade HTTP	Choose Yes to enable HTTP upgrade and provisioning. In “Check for new firmware every” field, enter the number of days to check the HTTP server for firmware upgrade or configuration.
Subscribe for MWI:	Default is NO. When set to Yes a SUBSCRIBE for Message Waiting Indication will be sent periodically.

Offhook Auto-Dial	This parameter allows a user to configure a User ID or extension number to be automatically dialed upon offhook. Please note that only the user part of a SIP address needs to be entered here. The HandyTone ATA will automatically append the “@” and the host portion of the corresponding SIP address. Note: Please write down the IP address of the ATA if you use this feature as it will disable the IVR and the only way to access it is via web configuration page.
Enable Call Features	Default is Yes. Call Forwarding & Do-Not-Disturb are supported locally
Disable Call Waiting	Default is No. User can use * code to use this feature per call basis.
Send DTMF	This parameter controls the way DTMF events are transmitted. There are 3 ways: in audio which means DTMF is combined in audio signal (not very reliable with low-bit-rate codec), via RTP (RFC2833), or via SIP INFO.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833
Send Flash Event	This parameter allows a user to control whether to send an SIP NOTIFY message indicating the Flash event, or just to switch to the voice channel when the user presses the Flash key. Default is NO.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port.
Caller ID Scheme	Select the Caller ID Scheme to suit the standard of different area.
Onhook Voltage	Select the onhook voltage to suit different area or PBX.
Polarity Reversal	Select Polarity Reversal to adapt some call charge/billing system. Default is NO.
NTP server	This parameter defines the URI or IP address of the NTP (Network Time Protocol) server which the HandyTone ATA will use to display the current date/time.
Send Anonymous	If this parameter is set to “Yes”, the “From” header in outgoing INVITE message will be set to anonymous, essentially block the Caller ID from displaying.
Lock keypad update	If this parameter is set to “Yes”, the configuration update via keypad is disabled.
Special Features	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.
WAN side http access	Default is “No”. The access to configuration page via WAN port is disabled. Need to change to “Yes” if user wants WAN side HTTP access to the ATA. If in Bridge mode, the http access is automatically enabled.
PSTN Access Code	Default is “*00”. By press the code, user can switch the phone connected to ATA to PSTN line connected to the Line port of ATA and call out using PSTN. User can change this to other access code like “8” or “9” or whatever user prefer. This is very useful when emergence happens to call “911”.
Syslog Server	The IP address or URL of System log server, especially useful for ITSP (Internet Telephone Service Provider)
Syslog Level	Select the ATA to report the log level. Default is NONE. Useful for ITSP.

6.2.4 Saving the Configuration Changes

Once a change is made, the user should click the “Update” button in the Configuration Menu. The HandyTone ATA will then display the following screen to confirm that the changes have been saved.



6.2.5 Rebooting the HandyTone-486 from remote

The user/administrator of the HandyTone-486 can remotely reboot the HandyTone ATA by pressing the “Reboot” button at the bottom of the configuration page. Once done, the following screen will be displayed to indicate that rebooting is underway.



At this point, the user can relogin to the HandyTone ATA after waiting for about 30 seconds.

6.3 Configuration through a Central Server

Grandstream HandyTone ATAs can be automatically configured from a central provisioning system.

When HandyTone ATA boot up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is “cfg.txt” and the other is “cfg000b82xxxxxx”, where “000b82xxxxxx” is the MAC address of the HandyTone ATA. For more information regarding provisioning configuration file format, please contact Grandstream and refer to the related Grandstream documentation.

The configuration file can be downloaded via TFTP or HTTP from the central provisioning server. A service provider or an enterprise with large deployment of HandyTone ATAs can easily manage the configuration and service provisioned to individual devices remotely and automatically from a central server. GAPS (Grandstream Automated Provisioning System) uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual HandyTone ATA for firmware upgrade, remote reboot, configuration changes, etc.

Grandstream provides a licensed provisioning system called GAPS that can be used to support automated configuration of HandyTone ATA. To enable this feature on the HandyTone ATA, a user just needs to enter the IP address of the GAPS server in the TFTP server field of the configuration screen, or enter the HTTP provisioning Server URL in the HTTP Upgrade Server field. Then reboot the HandyTone ATA.

For details on how GAPS works, please contact Grandstream and refer to the documentation of GAPS product provided.

7 Software Upgrade

7.1 Upgrade through HTTP

To upgrade software, HandyTone ATAs can be configured with an HTTP server where the new code image file is located. For example, following URL in the HTTP Upgrade Server:

<http://firmware.mycompany.com:6688/Grandstream/1.0.6.2>

Here, `firmware.mycompany.com` is the FQDN of the HTTP server, “:6688” is the TCP port the HTTP server listening to, “/Grandstream/1.0.6.2” is the RELATIVE directory to the root dir in HTTP web server. Thus, you can put different firmware into different directory as well.

NOTES:

- *To enable HTTP firmware upgrade, “Auto Upgrade” field has to be set to Yes. In addition, the ATA will check the HTTP server in the number of days that is defined in “Check for new firmware every” field.*

7.2 Upgrade through TFTP

To upgrade software, HandyTone ATAs can be configured with a TFTP server where the new code image is located. The TFTP upgrade can work in either static IP or DHCP mode using private or public IP address. It is recommended to set the TFTP server address either in a public IP address or on the same LAN with the HandyTone ATA.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt IVR or via the HandyTone ATA’s Web configuration interface. To configure the TFTP server via IVR and key pad, follow section 5.1 with option 06, once set up the TFTP IP address, power cycle the ATA, the firmware will be fetched once the ATA boots up.

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the HandyTone ATA. Input the admin password to enter the configuration screen. From there, enter the TFTP server address in the designated field towards the bottom of the configuration screen.

Once the TFTP server is configured, power cycle the HandyTone ATA to start TFTP upgrade.

TFTP checking is only performed during the initial power up. If the configured TFTP server is found and a new code image is available, the HandyTone ATA will attempt to retrieve the new image files by downloading them into the HandyTone ATA’s SRAM. During this stage, the HandyTone ATA’s LEDs will blink in red until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP fails for any reason (e.g., TFTP server is not responding, there are no code image files available for upgrade, or checksum test

fails, etc), the HandyTone ATA will stop the TFTP process and simply boot using the existing code image in the flash.

TFTP may take as long as 1 to 20 minutes depends on the networks' connection to Internet, or just 20+ seconds if it is performed on a LAN environment. It is generally recommended to conduct TFTP upgrade process in a controlled LAN environment if possible. For users who do not have local TFTP server, Grandstream provides a NAT-friendly TFTP server on the public Internet, also an HTTP web server (in the near future) for users to download the latest firmware and upgrade automatically. Please check the Service or Support section of Grandstream's Web site to obtain those servers' IP addresses.

8 Restore Factory Default Setting

Warning:

Restore the Factory Default Setting will erase all configuration information of the device.

- **Step one:** Find the MAC Address of the device. The MAC address of the device is located on the bottom of the device. It is a 12 digits hex number.
- **Step two:** Encode the MAC address to decimal digits. The encode rule is:
 - “2” is the first letter on the button “2” so its encoding is “2”.
 - “A” is the second letter on button “2” so its encoding is “22”.
 - “B” is the third letter on button “2” and its encoding is “222”.
 - “C” is the fourth letter on button “2” and its encoding is “2222”.

For example, for MAC address 000b8200e395, the user should encode it as “0002228200333395”.

- **Step three:** Access the voice menu, then dial “99” and get the voice prompt “RESET”
- **Step four:** Key in the encoded MAC address decimal digits after hear the IVR prompt. Once the correct encoded MAC address is keyed in, the device will reboot automatically and restore the factory default setting.

NOTES:

- *Be advised by default the HT486 WAN side HTTP access is disabled. After the factory reset, the **ONLY** way to get access to the configuration page is connecting your computer to the LAN port of HT486.*